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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

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Office Action Summary	Application No. 10/550,288	Applicant(s) MOGI ET AL.	
	Examiner Jean B. Jeanglaude	Art Unit 2819	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *amendment filed on 1-04-08*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-57* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-57* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☒ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

Response To Amendments/Arguments

Applicant's arguments with respect to claims 1 - 57 have been considered but are moot in view of the new ground(s) of rejection.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

1. Claims 1 - 14, 29 – 55, 57 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hellberg (US Patent Number 6,462,682) in view of Wu et al. (US PGPUB 2003/0161486).

2. Regarding claims 1, 46, Hellberg discloses a sampling rate converter and method (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold , a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse

response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

3. Regarding claims 6, 47, Hellberg discloses a sampling rate converter (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold, a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples

with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes a filter coefficient, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

4. Regarding claims 10, 48, Hellberg discloses a sampling rate converter and method (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold, a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined

convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

5. Regarding claims 29, 34, 49, Hellberg discloses a sampling rate converter (figs. 1 - 18) comprising: a convolution processing unit (12, fig. 18) including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and a linear interpolation block for finding the value at the required position from linear interpolation, wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 41; col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030,

0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

6. Regarding claims 35, 50, Hellberg discloses a sampling rate converter (figs. 1 – 18) comprising: a convolution processing unit (12, fig. 18) including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and a linear interpolation block for finding the value at the required position from linear interpolation, wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point (col. 1, lines 29 – 41; col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering

signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

7. Regarding claims 40, 45, 51 – 54, Hellberg discloses a sampling rate converter and method (figs. 1 – 18) comprising: a convolution processing unit (12, 18) including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67),, a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and a linear interpolation block for finding the value at the required position from linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter (col. 1, lines 29 – 41; col. 1, lines

29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

8. Regarding claim 55, Hellberg discloses an audio apparatus including a sampling rate converter (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold , a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse

response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

9. Regarding claim 57, Hellberg discloses an audio apparatus including sampling rate converter (figs. 1 – 18) comprising: a convolution processing unit (12, 18) including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase filter having a selected coefficient (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), a selector for selecting two points of samples required

for an output sample and selecting the coefficient of the corresponding poly-phase filter, and a linear interpolation block for finding the value at the required position from linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to frequency points to be passed and a frequency response of the pre-filter (col. 1, lines 29 – 41; col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg dos not specifically discloses a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

10. Regarding claims 2, 10, 11, 30, 36, 41, Hellberg discloses a sampling rate converter (figs. 1 – 18) wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with

respect to a desired characteristic in relation to a frequency response of the pre-filter col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

11. Regarding claims 3, 7, 12, 31, 37, 42, Hellberg discloses a sampling rate converter (figs. 1 – 18) wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

12. Regarding claims 4, 8, 13, 32, 38, 43, Hellberg discloses a sampling rate converter (figs. 1 – 18), further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower than a sampling frequency of the output (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

13. Regarding claims 5, 9, 14, 33, 39, 44, Hellberg discloses a sampling rate converter (figs. 1 – 17), further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

Claim Rejections - 35 USC § 103.

14. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action.

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention

was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

15. Claims 15 – 28, 56 are rejected under 35 U.S.C. 103(a) as being unpatentable over Hellberg (US Patent Number 6,462,682) Hellberg (US Patent Number 6,462,682) in view of Venkitachalam et al. (US Patent Number 6,542,094) and Wu et al. (US PGPUB 2003/0161486).

16. Regarding claim 15, Hellberg discloses a sampling rate converter (figs. 1 – 18) comprising: a plurality of convolution processing units (12, fig. 18) including pre-phase filters obtained by poly-phase decomposing a predetermined FIR filter and performing the convolution processing of input sample signals and the poly-phase filters decomposed to the poly-phases, corresponding convolution processing units and raising the sampling frequency U-fold(paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67); a linear interpolation block for selecting two points of samples with respect to the signal by the adding means and finding the value at the required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes the filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 41; col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg does not

specifically disclose a sampling rate converter that comprises a plurality of up-samplers. However, Venkitachalam et al., in a related field, discloses a sample rate converter (figs. 1 –9) that comprises a plurality of up-samplers (col. 12, lines 5 - 7). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Venkitachalam et al.'s system in order to improve the performance of the sample rate conversion and the combination of Hellberg and Venkitachalam et al. would achieve the same end result as the claimed invention.

17. Moreover both Hellberg and Venkitachalam et al. do not specifically disclose a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system combined with Venkitachalam et al.'s system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

18. Regarding claim 20, Hellberg discloses a sampling rate converter (figs. 1 – 18) comprising: a convolution processing unit (12, fig. 18) including poly-phase filters able to set different filter coefficients obtained by poly-phase decomposing a predetermined FIR filter and performing convolution processing of input sample signals and a poly-phase

filter having a selected coefficient (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), a selector for selecting two points of samples required for an output sample and selecting the coefficient of the corresponding poly-phase filter, and a linear interpolation block for finding the value at the required position from linear interpolation, wherein the FIR filter is an FIR filter where an impulse response is expressed by a finite time length, and the impulse response becomes the filter coefficient, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic using an algorithm adding a restrictive condition so as to pass any frequency point (col. 1, lines 29 – 41; col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg does not specifically disclose a sampling rate converter that comprises a plurality of up-samplers. However, Venkitachalam et al., in a related field, discloses a sample rate converter (figs. 1 –9) that comprises a plurality of up-samplers (col. 12, lines 5 - 7). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Venkitachalam et al.'s system in order to improve the performance of the sample rate conversion and the combination of Hellberg and Venkitachalam et al. would achieve the same end result as the claimed invention.

19. Moreover both Hellberg and Venkitachalam et al. do not specifically disclose a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear

interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system combined with Venkitachalam et al.'s system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

20. Regarding claim 55, Hellberg discloses a sampling rate converter (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-fold , a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49;

paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg does not specifically disclose a sampling rate converter that comprises a plurality of up-samplers. However, Venkitachalam et al., in a related field, discloses a sample rate converter (figs. 1 –9) that comprises a plurality of up-samplers (col. 12, lines 5 - 7). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Venkitachalam et al.'s system in order to improve the performance of the sample rate conversion and the combination of Hellberg and Venkitachalam et al. would achieve the same end result as the claimed invention.

21. Moreover both Hellberg and Venkitachalam et al. do not specifically disclose a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system combined with Venkitachalam et al.'s system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

22. Regarding claim 56, Hellberg discloses an audio apparatus including a sampling rate converter (figs. 1 – 18) comprising: an up sampler (24, fig. 18) for inserting U-1 zero points between sample signals and raising a sampling frequency U-

fold , a convolution processing unit (12, fig. 18) including an FIR filter and performing predetermined convolution processing with respect to an output signal of the up sampler (paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, lines 43 – 58; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67), and a linear interpolation block for selecting two points of samples with respect to the results of processing of the convolution processing unit and finding a value at a required position from the linear interpolation (col. 1, lines 29 – 41), wherein the FIR filter of the convolution processing unit is an FIR filter where an impulse response is expressed by a finite time length, the impulse response becomes a filter coefficient, and a transmission function H(z) is associated with a transmission function Z(z) of a pre-filter, and the filter coefficient is set by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter (col. 1, lines 29 – 65; paragraph bridging col. 1 and 2; col. 3, lines 25 – 49; paragraph bridging col. 3 and 4; paragraph bridging col. 4 and 5; col. 6, line 43 to col. 7, line 41; col. 10, lines 34 - 67). Hellberg does not specifically disclose a sampling rate converter that comprises a plurality of up-samplers. However, Venkitachalam et al., in a related field, discloses a sample rate converter (figs. 1 –9) that comprises a plurality of up-samplers (col. 12, lines 5 - 7). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system with that of Venkitachalam et al.'s system in order to improve the performance of the sample rate conversion and the combination of Hellberg and Venkitachalam et al. would achieve the same end result as the claimed invention.

23. Moreover both Hellberg and Venkitachalam et al. do not specifically disclose a sampling rate converter wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation. However, WU et al., in a related field, discloses a system (fig. 2) wherein a low pass filter providing either low pass filtered sample signals to the up sampler or low pass filtering signals output of the linear interpolation (abstract; paragraphs 0030, 0032). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Hellberg's system combined with Venkitachalam et al.'s system with that of Wu et al. in order to convert a digital audio signal from a non-standard sampling rate to any plurality of standard sampling rates.

24. Regarding claims 16, 25, Hellberg discloses a sampling rate converter (figs. 1 – 18) wherein the filter coefficient is set based on an amplitude characteristic of an equalizer obtained by performing weighted approximation with respect to a desired characteristic in relation to a frequency response of the pre-filter col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

25. Regarding claims 17, 21, 26, Hellberg discloses a sampling rate converter (figs. 1 – 18) wherein the weighted approximation is performed with respect to a desired characteristic using a Remex Exchange algorithm considering a frequency response of the pre-filter (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

26. Regarding claims 18, 22, 27, Hellberg discloses a sampling rate converter (figs. 1 – 18), further including a low bandpass filter preventing an aliasing component from occurring and folding from occurring when the sampling frequency of the input is lower

than a sampling frequency of the output (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

27. Regarding claims 19, 23, 28, Hellberg discloses a sampling rate converter (figs. 1 – 17), further including a low bandpass filter preventing an imaging component from occurring and a non-original frequency component from occurring when the sampling frequency of the input is higher than a sampling frequency of the output (col. 6, line 43 to col. 7, line 41; col. 10, lines 34 – 67).

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. (See PTO-892).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jean B. Jeanglaude whose telephone number is 571-272-1804. The examiner can normally be reached on Monday - Friday 7:30 A. M. - 5:00 P.M..

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Rexford Barnie can be reached on 571-272-7492. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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